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| APPLICATION NO. | FILING DATE | FIRST NAMED INVENTOR | ATTORNEY DOCKET NO. | CONFIRMATION NO. |
|---|---|---|---|---|
| 10/725,294 | 12/01/2003 | Toru Marumoto | 9333/360 | 2919 |

74989 7590 12/24/2009

ALPINE/BHGL
P.O. Box 10395
Chicago, IL 60610

| EXAMINER | |
|---|---|
| COLUCCI, MICHAEL C | |
| **ART UNIT** | **PAPER NUMBER** |
| 2626 | |
| **MAIL DATE** | **DELIVERY MODE** |
| 12/24/2009 | PAPER |

**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/725,294 | MARUMOTO ET AL. | |
| | Examiner | Art Unit | |
| | MICHAEL C. COLUCCI | 2626 | |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on *18 November 2009*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-3,5,7-11,13-17 and 21* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-3,5,7-11,13-17 and 21* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

## DETAILED ACTION

### *Response to Arguments*

1. Applicant's arguments filed 11/18/2009 have been fully considered but they are not persuasive.

**Argument (page 10 paragraph 1):**

- "For at least this reason, Andreas and Soli, alone or in combination, do not teach a speech communication apparatus comprising received-speech clarifying means for adjusting a gain for a received-speech signal based on a power level of a background sound measured by a background sound level measurement means, wherein the received speech signal is transmitted to the speech communication apparatus for output by a speaker and comprises speech that is not received from a user of the speech communication apparatus as asserted by the Examiner"

**Response to argument:**

**NOTE:** Examiner would like to remind Applicant of the following:

*"USPTO personnel are to give claims their broadest reasonable interpretation in light of the supporting disclosure. In re Morris, 127 F.3d 1048, 1054-55, 44 USPQ2d 1023,1027-28 (Fed. Cir. 1997). Limitations appearing in the specification but not recited in the claim should not be read into the claim. E-Pass Techs., Inc. v. 3Com Corp., 343 F.3d1364, 1369, 67 USPQ2d 1947, 1950 (Fed.*

*Cir. 2003) (claims must be interpreted "in view of the specification" without importing limitations from the specification into the claims unnecessarily). In re Prater, 415 F.2d 1393, 1404-05, 162 USPQ 541, 550-551 (CCPA 1969). See also In re Zletz, 893 F.2d 319, 321-22, 13 USPQ2d 1320, 1322 (Fed. Cir. 1989) ("During patent examination the pending claims must be interpreted as broadly as their terms reasonably allow.... The reason is simply that during patent prosecution when claims can be amended, ambiguities should be recognized, scope and breadth of language explored, and clarification imposed.... An essential purpose of patent examination is to fashion claims that are precise, clear, correct, and unambiguous. Only in this way can uncertainties of claim scope be removed, as much as possible, during the administrative process."). Where an explicit definition is provided by the applicant for a term, that definition will control interpretation of the term as it is used in the claim. Toro Co. v. White Consolidated Industries Inc., 199 F.3d 1295, 1301, 53 USPQ2d 1065, 1069 (Fed. Cir. 1999) (meaning of words used in a claim is not construed in a "lexicographic vacuum, but in the context of the specification and drawings."). Any special meaning assigned to a term "must be sufficiently clear in the specification that any departure from common usage would be so understood by a person of experience in the field of the invention." Multiform Desiccants Inc. v. Medzam Ltd., 133 F.3d 1473, 1477, 45 USPQ2d 1429, 1432 (Fed. Cir. 1998). See also MPEP § 2111.01."*

Examiner maintains the use of Andrea in view of Soli and Suzuki while giving claims their broadest reasonable interpretation in light of the supporting disclosure without importing limitations from the specification into the claims unnecessarily, wherein Andrea alone is believed to teach "a speech communication apparatus comprising received-speech clarifying means for adjusting a gain for a received-speech signal based on a power level of a background sound measured by a background sound level measurement means, wherein the received speech signal is transmitted to the speech communication apparatus for output by a speaker and comprises speech that is not received from a user of the speech communication apparatus".

Please consider the following analysis of the claim language in view of Andrea:

- "a speech communication apparatus comprising received-speech clarifying means"

It is well known in the art that an OP-AMP controls the gain of a signal, where Andrea teaches that <u>an electrical signal is supplied representing substantially the speech to the telephone unit 18 whereupon the speech signal is transmitted therefrom through the telephone lines to a desired telephone or telephones</u>. The <u>output signal from the op-amp 16 is also combined in the telephone unit 18 with a received signal from the telephone lines and supplied to the amplifier 20</u>. Although not shown in FIG. 2, amplifier 20 may be selectively set by use of the switch 40 (FIG. 1) by the operator so as to

adjust the amplification of the received signal to a desired level. The amplified signal from the amplifier 20 is supplied to the speaker 22, whereupon the amplified signal is converted into an acoustic signal so as to be heard by the operator (Andrea Col. 12 line 55 – Col. 13 line 14 & Fig. 2 and 28).

- "clarifying means for adjusting a gain for a received-speech signal based on a power level of a background sound measured by a background sound level measurement means"

Andrea teaches an active noise reduction system, where B is the variable gain/phase controller or calibration pot 1350; H.sub.1 represents the high pass filter 1380 and N.sub.1 is the noise 1390 at the quiet zone at the ear of the user. The active noise reduction system is comprised of an open loop circuit having the following components: an audio signal 1340; a sensor microphone 1310 able to detect and cancel noise 1315; a output transducer 1370 located near the user's ear; a variable gain/phase controller 1350 to adjust the amplitude of the anti-noise 1315 (not noise but speech); a summing node 1360 to sum the anti-noise signal and audio signal 1340; a high pass filter 1380 to prevent mechanical vibration induced frequency disturbance components 1320 from reaching the output transducer. The system detects ambient noise 1315 by the sensor or pickup microphone and applies electroacoustical processing to produce an acoustical signal for canceling out the ambient noise. This system may be used to cancel all

noise, so as to obtain a signal representing speech, which is the desired signal to be heard through the ear of the user (Andrea Col. 29 line 59 – Col. 13 line 12).

Further and more specifically specific, consider that Andrea teaches that when the voice and noise microphone 1901 is enabled, the microphone amplifier 1940 output at A 1905 is the noise omnidirectional microphone 1901 output. In the talk-thru mode, the VOX circuit 1950 is bypassed by the second S1B switch 1925 placed in the talk thru position, which allows the direct output of the noise omnidirectional microphone signal 1901 to the buffer amplifier 1935 at D 1903 and then outputted to the audio system 1915 at E. As a result, no speech signal is inputted to transmission gate 1945 output at B 1904 because the gate 1945 is disabled. The gain control function 1907 of the scaling amplifier 1970 is increased at W 1906, by the action of switch S1C 1960, at the talk thru position 1930. Thus, the sidetone signal outputted from the active noise reduction system (described in FIGS. 21-27) is increased at the speaker 1980 (Andrea Col. 34 lines 39-55 & Fig 28).

Similarly, when the system is in the noise canceling mode, the switch S1 is in the "N" position 1910, and the active noise cancellation microphones 1900 and 1901 are operating as previous described herein. When the VOX 1950 has determined that the signal at the output of the microphone is useable audio, it will activate the "speech signal" at point C 1955, which will enable the transmission gate 1945 and allow the microphone audio outputted from the active noise cancellation microphone amplifier

means 1940 into the buffer amplifier 1935 and then to the audio system at E 1915. In addition, this audio 1915 is sent to the scaling amplifier 1970 at W 1906 to provide a sidetone signal to the earcup speakers 1980 when the third switch S1C 1960 is at N, the noise canceling mode. The scaling amplifier 1970 can also simultaneously accept an input from an external audio system 1990, i.e. distinctive sounds from the surroundings, such as sirens, bystander's voices, or other external sounds not being transmitted by the microphones 1900 and 1901. The composite signal at the earcup speaker 1980 is the linear addition of sidetone 1960 and external audio 1990 (Andrea Col. 34 lines 10-38).

In other words (in view of Fig. 28), when the single microphone 1901 is used noise and speech are sent into the gain combination of scaling amp 1970 and gain control 1907, the output 1980 is altered based on input speech (or speech and noise). Further, "adjusting a gain for a received-speech signal based on a power level of a background sound", particularly "*based on a power level of a background sound"*, while giving claims their broadest reasonable interpretation in light of the supporting disclosure without importing limitations from the specification into the claims unnecessarily, appears to be taught by Andrea, wherein the received speech signal gain of a first user/caller (1901 or 1900) is altered (output E) prior to being output to a speaker 1980 for a second speaker. For example, consider voice signal 1900 gain adjustment based on background noise input from 1990. Gain adjustment is present "*based on a power level of a background sound",* where power or even voltage is demonstrated by, for example the frequency

response of Andrea's Fig. 13A, wherein a decibel is merely a logarithmic unit of measurement that expresses the magnitude of a physical quantity such as power or intensity. The scaling of an input signal E having a reference signal (1990) provides a clear output to the ear of another user.

Examiner would also like to point out a the improvement of Andrea that can be achieved via Soli, wherein Soli teaches the use of a reference noise signal, where a single microphone is used to receive both wanted and unwanted parts of the auditory signal and the total auditory signal is processed to de-emphasize the unwanted part, i.e., the noise, relative to the wanted part, i.e., the speech. For example, a good deal of unwanted noise usually exists in the low frequency bands of speech and can actually mask some of the desired high frequency parts of speech. (This is called the upward spread of masking.) By de-emphasizing the lower frequency parts of the signal, i.e. ., attenuating the frequencies between 50 and 500 Hertz, for example, the unwanted noise signal is decreased (along with some of the wanted speech signal) making the higher frequency parts of the speech discernible. The overall effect can be to increase the intelligibility of speech in the presence of noise. One variation of the single microphone technique is to provide a directionality to the microphone so that the wearer (user) can optimize the wanted part of the signal, the speech, while decreasing any unwanted part of the signal, the noise, which is not directionally coincident with the speech signal (Soli Col. 2 lines 1-20, & Fig. 1 amplifier 13 amplifying the input signal sent to speaker 33 of Fig. 2).

- "the received speech signal is transmitted to the speech communication apparatus for output by a speaker and comprises speech that is not received from a user of the speech communication apparatus"

Andrea teaches a two user system, wherein the method for reducing noise according to this invention is provided by an open loop circuit allowing the input audio signal from an operator or caller to be transmitted to the user's ear without the disturbance of unwanted ambient noise (Andrea Abstract).

Applicant is encouraged, if necessary, to schedule an interview to propose amendments to overcome Examiners prior art of record and continue prosecution efficiently.

### *Claim Rejections - 35 USC § 103*

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1, 3, 5, 7, 8, 9, and 21 are rejected under 35 U.S.C. 103(a) as being unpatentable over Andrea et al. US 5732143 A (hereinafter Andrea) in view of Soli et al. US 6563931 B1 (hereinafter Soli) and further in view of Suzuki US 4420655 A (hereinafter Suzuki).

Re claims 1 and 21, Andrea teaches a speech communication apparatus for bi-directional speech communications, comprising:

a speaker (Fig. 1 speaker 22);

a microphone (Fig. 1 Mic 12);

transmission means for transmitting speech to be transmitted which has been extracted by the transmission-speech signal generation filter (Fig. 2, input from telephone lines to unit 18)

received-speech clarifying means for adjusting a gain for a received-speech signal to be output by the speaker based on the level of the background sound (Col. 34 lines 10-38) from the output signal measured by the background sound level measurement means (Fig. 2, AMP 20, adjusting speech from AMP 16 and from telephone lines via unit 18);

wherein the received speech signal is transmitted to the speech communication apparatus for output by the speaker and comprises speech that is not received from a user of the speech communication apparatus (Fig. 2, received speech at unit 18 is received separately from the speech at MIC 12).

However, Andrea fails to teach two signals together, wherein one signal is the received speech and the other signal is the background sound level.

wherein the speech communication apparatus does not comprise more than one microphone

Soli teaches a single microphone technique that is used to provide a directionality to the microphone so that the wearer (user) can optimize the wanted part of the signal, the speech, while decreasing any unwanted part of the signal, the noise, which is not directionally coincident with the speech signal (Soli Col. 2 lines 16-20 & Fig. 1 items 6, 8, and 11).

Additionally, Soli teaches a single microphone 11 providing the noise reference signal and primary input signal, the adaptive filter 16 will tend to cancel desired signal as well as noise if desired signal is present in the input signal while filter 16 is adapting (Soli Col. 7 lines 45-56). The present invention thus allows that a human, such as the user, actuate the adapting mode of the filter 10 when noise alone is present at the microphone, to the best extent possible. For example, the user could wait for a pause in a conversation, or request a pause in a conversation, and actuate the adapting mode during this pause. This allows filter 10 to adapt to characteristics minimizing the noise passing through the filter without causing loss of the desired signal

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea to incorporate two signals, wherein one signal is the received speech and the other signal is the background sound level as taught by Soli to allow for a system that adapts in order to minimize noise and preserve

the quality of a received signal, wherein a user can intervene to enable the adaptation to maximize the received signal (i.e. hearing aids, phone, etc.) (Soli Col. 7 lines 45-56).

However, Andrea in view of Soli fails to teach a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal;

a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter

background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone

Suzuki teaches a circuit for compensating for frequency characteristic of microphone output which is arranged to have a combination of a microphone member of the pressure gradient type having a proximity effect represented by a rise in its low frequency range sensitivity as the microphone member approaches closer to a source of sound, and another microphone member of the pressure type developing no proximity effect, to cancel out the occurrence of proximity effect. A change in sensitivity of the pressure gradient type microphone in the low frequency range due to the proximity effect is determined from the level difference between the outputs of the two microphones. A signal representative of the difference is subtracted from the output of

the pressure gradient type microphone thereby to effect a compensation for the proximity effect in the output of the pressure gradient type microphone (Suzuki Abstract & fig. 3).

Further, Suzuki teaches well known uses of a filter used to reduce the proximity effect by applying a frequency scale based on the proximity effect onto a signal (Suzuki Fig. 2,4, and 6), wherein the difference between the output of a microphone and the output of a filter are generated with respect to the proximity effect (Suzuki Fig. 1).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea in view of Soli to incorporate a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal, a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter, and background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone as taught by Suzuki to allow for noise reduction within a greater frequency range, wherein frequency response is independent of the minimization of the proximity effect, and can therefore reduce noise within the whole frequency range (Suzuki Col. 3

lines 38-52 & Fig. 3) whereby the proximity effect is compensated for through a subtractive filter means (Suzuki Fig. 1).

Re claims 3 and 7, Andrea teaches the speech communication apparatus of claim 1, wherein the speech communication apparatus is a portable, mobile telephone for performing the speech communications by radio communication (Col. 12 lines 35-45, cordless RF transmission).

Re claim 5, Andrea teaches the speech communication apparatus of claim 1, wherein the microphone is a unidirectional or bi-directional microphone (Col. 3 lines 18-50).

Re claim 8, Andrea teaches a speech communication apparatus comprising:

a speaker (Fig. 1 speaker 22);

a microphone (Fig. 1 Mic 12);

a background sound microphone (Fig. 1 Mic 14);

a background sound level calculator operable to calculate a level of a signal outputted from the adder and a level of the background sound (Fig. 2 Op Amp 16 difference of speech and noise inputs);

a received speech clarifying filter operable to adjust a gain for received speech to be output by the speaker based on the background sound level, wherein the received speech is transmitted to the speech communication apparatus for output by the speaker

and comprises speech that is not received from a user of the speech communication apparatus (Fig. 2, received speech at unit 18 is received separately from the speech at MIC 12).

However, Andrea fails to teach a transmission speech filter operable to reduce a level of a lower frequency component of an output signal from the microphone (Soli Col. 2 lines 1-15);

a background sound level filter operable to minimize proximity effect;

an adaptive filter operable to estimate speech signals from the background sound microphone (Soli Fig. 1 item 16);

an adder operable to subtract the estimated speech signal from the output of the background sound microphone (Soli Fig. 1 item 17);

two signals, wherein one signal is the received speech and the other signal is the background sound level.

Soli teaches a single microphone technique that is used to provide a directionality to the microphone so that the wearer (user) can optimize the wanted part of the signal, the speech, while decreasing any unwanted part of the signal, the noise, which is not directionally coincident with the speech signal (Soli Col. 2 lines 16-20 & Fig. 1 items 6, 8, and 11).

Additionally, Soli teaches a single microphone 11 providing the noise reference signal and primary input signal, the adaptive filter 16 will tend to cancel desired signal as well as noise if desired signal is present in the input signal while filter 16 is adapting

(Soli Col. 7 lines 45-56). The present invention thus allows that a human, such as the user, actuate the adapting mode of the filter 10 when noise alone is present at the microphone, to the best extent possible. For example, the user could wait for a pause in a conversation, or request a pause in a conversation, and actuate the adapting mode during this pause. This allows filter 10 to adapt to characteristics minimizing the noise passing through the filter without causing loss of the desired signal

However, Andrea in view of Soli fails to teach the minimization of the proximity effect.

Suzuki teaches a circuit for compensating for frequency characteristic of microphone output which is arranged to have a combination of a microphone member of the pressure gradient type having a proximity effect represented by a rise in its low frequency range sensitivity as the microphone member approaches closer to a source of sound, and another microphone member of the pressure type developing no proximity effect, to cancel out the occurrence of proximity effect (Suzuki Abstract & fig. 3).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea to incorporate two signals, wherein one signal is the received speech and the other signal is the background sound level as taught by Soli to allow for a system that adapts in order to minimize noise and preserve the quality of a received signal, wherein a user can intervene to enable the adaptation to maximize the received signal (i.e. hearing aids, phone, etc.) (Soli Col. 7 lines 45-56).

Additionally, it would also have been obvious to one of ordinary skill in the art at the time of the invention to incorporate the minimization of the proximity effect as taught by Andrea in view of Soli to allow for noise reduction within a greater frequency range, wherein frequency response is independent of the minimization of the proximity effect, and can therefore reduce noise within the whole frequency range (Suzuki Col. 3 lines 38-52 & Fig. 3).

Re claim 9, Andrea teaches the speech communication apparatus of claim 8, further comprising:

transmission means for transmitting an output of the transmission-speech filter as a transmission-speech signal from the speech communications apparatus (Fig. 2 input/output at telephone unit 18).

**4. Claim 2 are rejected under 35 U.S.C. 103(a) as being unpatentable over Andrea et al. US 5732143 A (hereinafter Andrea) in view of Soli et al. US 6563931 B1 (hereinafter Soli) and Suzuki US 4420655 A (hereinafter Suzuki) and further in view of Urbanski, US 5,544,250 A (hereinafter Urbanski).**

Re claim 2, Andrea fails to teach the speech communication apparatus of claim 1, further comprising:

received-speech-level measurement means for measuring a level of the received-speech signal at each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3),

wherein the background sound level measurement means measures the level of the background sound in each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3) and the received-speech clarifying means performs loudness compensation in which the gain for the received-speech signal is adjusted (Fig. 2 item 207) in each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3) such that received speech output by the speaker is heard at almost the same intensity in the human auditory sense irrespective of the level of the background sound (Col. 1 lines 21-41 & Fig. 3), and the resultant signal is output to the speaker as the received speech (Col. 1 lines 42-56).

Urbanski teaches a microphone 101 is coupled to receive an input signal 113, in acoustic form, to produce the input signal in electric form at line 115. The A/D converter 103, operatively coupled to the microphone 101, converts the input signal in electric form at line 115 to a digital signal at line 116. The noise suppression system 105, operatively coupled to the A/D converter 103, produces a noise suppressed signal at line 117 responsive to the input signal at line 116. The audio signal processor 107, operatively coupled to the noise suppression system 105, produces a processed signal at line 119 responsive to the noise suppressed signal at line 117. The transmitter 109, operatively coupled to the audio processor 107, transmits the processed signal at line 119 to produce a transmitted signal in electric form at line 121. The antenna 111, operatively coupled to the transmitter 109, converts the transmitted signal in electric

form at line 121 to a transmitted signal in electromagnetic form as represented by reference number 123. The communication unit 100 is preferably a cellular radiotelephone. For example, the cellular radiotelephone may be a digital cellular radiotelephone, such as used in the North American Digital Cellular (NADC) System; the Japan Digital Cellular (JDC) System; or the Group Special Mobile (GSM) System. Alternatively, the communication unit 100 may be a two-way radio, cordless radiotelephone, or a wireless microphone (Urbanski Col. 3 line 65 – Col. 4 line 4).

Further, Urbanski teaches acoustic noise suppression in a speech communication system generally serves the purpose of improving the overall quality of the desired audio signal by filtering environmental background noise from the desired speech signal. This speech enhancement process is particularly necessary in environments having abnormally high levels of ambient background noise, such as an aircraft, a moving vehicle, or a noisy factory. One noise suppression technique is a spectral subtraction--or a spectral gain modification--technique. Using this approach, the audio input signal is divided into individual spectral bands by a bank of bandpass filters, and particular spectral bands are attenuated according to their noise energy content. A spectral subtraction noise suppression prefilter utilizes an estimate of the background noise power spectral density to generate a signal-to-noise ratio (SNR) of the speech in each channel, which, in turn, is used to compute a gain factor for each individual channel. The gain factor is used as the attenuation for that particular spectral band. The channels are then attenuated and recombined to produce the noise-suppressed output waveform. In specialized applications involving relatively high

background noise environments, most noise suppression techniques exhibit significant performance limitations. One example of such an application is the vehicle speakerphone option to a cellular mobile radio telephone system, which provides hands-free operation for the automobile driver. The mobile hands-free microphone is typically located at a greater distance from the user, such as being mounted overhead on the visor. The more distant microphone delivers a much poorer signal-to-noise ratio to the land-end party due to road and wind noise conditions. Although the received speech at the land-end is usually intelligible, continuous exposure to such background noise levels often increases listener fatigue. In rapidly-changing high noise environments, a severe low frequency noise flutter develops in the output speech signal which resembles a distant "jet engine roar" sound. This noise flutter is inherent in a spectral subtraction noise suppression system, since the individual channel gain parameters are continuously being updated in response to the changing background noise environment (Urbanski Col. 1 lines 20-63 & Fig. 3 bands).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea in view of Soli and Suzuki to incorporate received-speech-level measurement means for measuring a level of the received-speech signal at each predetermined frequency band, wherein the background sound level measurement means measures the level of the background sound in each predetermined frequency band and the received-speech clarifying means performs loudness compensation in which the gain for the received-speech signal is adjusted in each predetermined frequency band such that received speech output by the speaker is

heard at almost the same intensity in the human auditory sense irrespective of the level of the background sound, and the resultant signal is output to the speaker as the received speech as taught by Urbanski to allow for gain parameters that are continuously being updated in response to the changing background noise environment in various frequency bands, wherein a user can operate said communication device in a noisy environment such as an automobile and still hear the speaker (Urbanski Col. 1 lines 20-63 & Fig. 3).

**5. Claims 10, 11, and 13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Andrea et al. US 5732143 A (hereinafter Andrea) in view of Suzuki US 4420655 A (hereinafter Suzuki) and further in view of Todter et al US 5937070 A (hereinafter Todter).**

Re claim 10, Andrea teaches a speech communication apparatus for bi-directional speech communications, provided with a handset having at a front face a speaker for outputting received speech (Fig. 1 element 41) and a transmission-speech microphone (Fig. 1 element 46) for collecting speech to be transmitted (Fig. 2 Mic 12), the speech communication apparatus comprising:

transmission means for transmitting speech to be transmitted which has been extracted by the transmission-speech signal generation filter (Fig. 2, AMP 20, adjusting speech from AMP 16 and from telephone lines via unit 18).

background sound level measurement means for measuring a level of an output from the background-sound microphone as a background-sound level (Fig. 2, OP AMP 16 from Mic 14);

received-speech clarifying means for adjusting a gain for received speech that is output from the speaker based on the background-sound level measured by the background sound level measurement means, wherein the received speech is transmitted to the speech communication apparatus for output by the speaker and comprises speech that is not received from a user of the speech communication apparatus (Fig. 2, received speech at unit 18 is received separately from the speech at MIC 12).

However, Andrea fails to teach a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal;

a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter

background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone

Suzuki teaches a circuit for compensating for frequency characteristic of microphone output which is arranged to have a combination of a microphone member of

the pressure gradient type having a proximity effect represented by a rise in its low frequency range sensitivity as the microphone member approaches closer to a source of sound, and another microphone member of the pressure type developing no proximity effect, to cancel out the occurrence of proximity effect. A change in sensitivity of the pressure gradient type microphone in the low frequency range due to the proximity effect is determined from the level difference between the outputs of the two microphones. A signal representative of the difference is subtracted from the output of the pressure gradient type microphone thereby to effect a compensation for the proximity effect in the output of the pressure gradient type microphone (Suzuki Abstract & fig. 3).

Further, Suzuki teaches well known uses of a filter used to reduce the proximity effect by applying a frequency scale based on the proximity effect onto a signal (Suzuki Fig. 2,4, and 6), wherein the difference between the output of a microphone and the output of a filter are generated with respect to the proximity effect (Suzuki Fig. 1).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea to incorporate a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal, a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the

transmission-speech signal generation filter, and background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone as taught by Suzuki to allow for noise reduction within a greater frequency range, wherein frequency response is independent of the minimization of the proximity effect, and can therefore reduce noise within the whole frequency range (Suzuki Col. 3 lines 38-52 & Fig. 3) whereby the proximity effect is compensated for through a subtractive filter means (Suzuki Fig. 1).

However, Andrea in view of Suzuki fails to teach a background-sound microphone disposed at the rear face of the handset at almost the same height as the speaker, for collecting background sound (Todter Col. 8 lines 50-65 & Col. 9 lines 12-25);

Todter teaches a plurality of pick ups or microphones are provided (e.g. in a telephone handset) at known positions in relation to the source of the desired audio signal, so that all microphones will receive the desired audio signal (albeit possibly with gain and phase differences) and all microphones will receive the extraneous noise signal (again possibly with gain and phase differences). The signals from each microphone may then be separately processed to provide electrical noise cancellation and added (possibly with appropriate weighting) to give the noise cancelled audio signal. The noise cancelling processing may make use of known propagation

characteristic differences between the ambient noise field and the desired audio signal acoustic field.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea in view of Suzuki to as taught by Todter to allow for a user of the headset to be able to communicate and transmit data in response to incoming data, wherein using a microphone one or more microphones a headset would allow for the acquisition of the same noise signal but with various gains and phase differences, where several noise cancellation operations can be performed and summed to produce a desirable speech signal (Todter Col. 8 lines 50-65 & Col. 9 lines 12-25).

Re claim 11, Andrea teaches the speech communication apparatus of claim 10, wherein the microphone is a unidirectional or bi-directional microphone (Col. 3 lines 18-50).

Re claim 13, Andrea teaches the speech communication apparatus of claim 1, wherein the speech communication apparatus is a portable, mobile telephone for performing the speech communications by radio communication (Col. 12 lines 35-45, cordless RF transmission).

**6. Claims 14, 15, and 17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Andrea et al. US 5732143 A (hereinafter Andrea) in view of Soli et al. US 6563931 B1 (hereinafter Soli) and further in view of Todter et al US 5937070 A (hereinafter Todter).**

Re claims 14, Andrea teaches a speech communication apparatus for bi-directional speech communications, comprising:

a speaker for outputting received speech (Fig. 1 speaker 22)

a microphone for collecting speech to be transmitted (Fig. 1 Mic 12);

background sound level measurement calculator operable to measure a level of background sound (Fig. 1 Mic 14);

a received-speech clarifying section operable to adjust a gain for the received speech to be outputted by the speaker based on the level of the background sound measured by the background sound level measurement calculator, wherein the received speech that is transmitted to the speech communication apparatus to be outputted by the speaker comprises speech that is not received from a user of the speech communication apparatus (Fig. 2, received speech at unit 18 is received separately from the speech at MIC 12).

However, Andrea in view of Suzuki fails to teach a delay section operable to delay an output of a first background-sound microphone by a period of time corresponding to a delay time between transmission speech mixed into the output of the

first background-sound microphone and transmission speech mixed into an output of a second background-sound microphone (Todter Col. 15 lines 1-15 & Fig. 12 item 72),

an adaptive filter (Todter Col. 13 lines 31-49) operable to estimate transmission of speech mixed into the output of the delay section (Todter Col. 15 lines 1-15 & Fig. 12 item 72),

an adder operable to subtract the transmission speech estimated by the adaptive filter from an output of the delay section (Todter Col. 15 lines 1-15 & Fig. 12 items 72 and 73),

a background sound level calculation section operable to calculate a level of an output of the adder and for outputting the result as the level of the background sound (Todter Col. 15 lines 1-15 & Fig. 12 items 74 and summation node prior to output just below item 74).

Todter teaches a high pass frequency segment that provides for a weighted and phase-shifted sum of the "n" microphone signals; containing both phase correlated speaker's voice and uncorrelated external noise signals. The invention allows for the adjustment of signal weighting and phase shifting to amplify the speakers voice signal and attenuate the external noise. (59) The high frequency sections comprises a high pass filter 80 and a plurality of gain and delay blocks 81, 82, respectively, as well as a plurality of positive summing circuits 83. The low pass frequency filter may also comprise a plurality of circuits 71, 72, 73. (60) The low pass frequency segment of the invention provides for a weighted and phase-shifted subtraction of noise from the mouthpiece microphone signal. The invention allows for adjustment of gain weighting

and phase shift to find the optimum improvement in signal to noise ratio, in any specific reverberant noise environment. (61) The block 60 further provides for the weighted summing via summing circuit 90 of low passed and high passed signals to reconstitute the total enhanced signal.

Additionally, Todter teaches an adaptive control block 9 that compares the noise cancellation signal output from the summing circuit 13 with the microphone signal to detect residual noise, and controls the gain compensation 20 and phase compensation 21 to keep the residual noise to a minimum. Also by negatively summing the audio signal into the adaptive control loop the loop will affect the compensation of the cancellation loop block to provide a high fidelity output signal, notwithstanding quality of components, as discussed in the preamble.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a delay section utilizing several microphones where delayed speech from several microphones are mixed together, using an adaptive filter in for the subtraction of speech from a delay section. Delaying segments of speech would be necessary to add high pass filtered and low pass filtered components, where low pass components can be weighted and summed in addition to high pass components, to reduce the amount of noise in a signal. In a phase shifting environment, summation of delayed components from several microphones would allow for reduced noise, adjustable gain, and optimal signal to noise ratio conditions in a noisy or non-noisy environment.

Re claim 15, Andrea in view of Soli fails to teach the speech communication apparatus of [[to]] claim 14, wherein the adaptive filter (Todter Col. 13 lines 31-49) estimates the transmission speech based on a difference between the output of the delay section and the transmission speech estimated by the adaptive filter (Todter Col. 15 lines 1-15 & Fig. 12 items 72 and 73).

Todter teaches a high pass frequency segment that provides for a weighted and phase-shifted sum of the "n" microphone signals; containing both phase correlated speaker's voice and uncorrelated external noise signals. The invention allows for the adjustment of signal weighting and phase shifting to amplify the speakers voice signal and attenuate the external noise. (59) The high frequency sections comprises a high pass filter 80 and a plurality of gain and delay blocks 81, 82, respectively, as well as a plurality of positive summing circuits 83. The low pass frequency filter may also comprise a plurality of circuits 71, 72, 73. (60) The low pass frequency segment of the invention provides for a weighted and phase-shifted subtraction of noise from the mouthpiece microphone signal. The invention allows for adjustment of gain weighting and phase shift to find the optimum improvement in signal to noise ratio, in any specific reverberant noise environment. (61) The block 60 further provides for the weighted summing via summing circuit 90 of low passed and high passed signals to reconstitute the total enhanced signal.

Additionally, Todter teaches an adaptive control block 9 that compares the noise cancellation signal output from the summing circuit 13 with the microphone signal to detect residual noise, and controls the gain compensation 20 and phase compensation

21 to keep the residual noise to a minimum. Also by negatively summing the audio signal into the adaptive control loop the loop will affect the compensation of the cancellation loop block to provide a high fidelity output signal, notwithstanding quality of components, as discussed in the preamble.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a delay section utilizing several microphones where delayed speech from several microphones are mixed together, using an adaptive filter in for the subtraction of speech from a delay section. Delaying segments of speech would be necessary to add high pass filtered and low pass filtered components, where low pass components can be weighted and summed in addition to high pass components, to reduce the amount of noise in a signal. In a phase shifting environment, summation of delayed components from several microphones would allow for reduced noise, adjustable gain, and optimal signal to noise ratio conditions in a noisy or non-noisy environment.

Re claim 17, Andrea teaches the speech communication apparatus of claim 1, wherein the speech communication apparatus is a portable, mobile telephone for performing the speech communications by radio communication (Col. 12 lines 35-45, cordless RF transmission).

**7. Claim 16 are rejected under 35 U.S.C. 103(a) as being unpatentable over Andrea et al. US 5732143 A (hereinafter Andrea) in view of Soli et al. US 6563931 B1 (hereinafter Soli) and Todter et al US 5937070 A (hereinafter Todter) and further in view of Urbanski, US 5,544,250 A (hereinafter Urbanski).**

Re claim 16, Andrea fails to teach the speech communication apparatus of claim 1, further comprising:

received-speech-level measurement means for measuring a level of the received-speech signal at each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3),

wherein the background sound level measurement means measures the level of the background sound in each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3) and the received-speech clarifying means performs loudness compensation in which the gain for the received-speech signal is adjusted (Fig. 2 item 207) in each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3) such that received speech output by the speaker is heard at almost the same intensity in the human auditory sense irrespective of the level of the background sound (Col. 1 lines 21-41 & Fig. 3), and the resultant signal is output to the speaker as the received speech (Col. 1 lines 42-56).

Urbanski teaches a microphone 101 is coupled to receive an input signal 113, in acoustic form, to produce the input signal in electric form at line 115. The A/D converter 103, operatively coupled to the microphone 101, converts the input signal in electric form at line 115 to a digital signal at line 116. The noise suppression system 105, operatively coupled to the A/D converter 103, produces a noise suppressed signal at

line 117 responsive to the input signal at line 116. The audio signal processor 107, operatively coupled to the noise suppression system 105, produces a processed signal at line 119 responsive to the noise suppressed signal at line 117. The transmitter 109, operatively coupled to the audio processor 107, transmits the processed signal at line 119 to produce a transmitted signal in electric form at line 121. The antenna 111, operatively coupled to the transmitter 109, converts the transmitted signal in electric form at line 121 to a transmitted signal in electromagnetic form as represented by reference number 123. The communication unit 100 is preferably a cellular radiotelephone. For example, the cellular radiotelephone may be a digital cellular radiotelephone, such as used in the North American Digital Cellular (NADC) System; the Japan Digital Cellular (JDC) System; or the Group Special Mobile (GSM) System. Alternatively, the communication unit 100 may be a two-way radio, cordless radiotelephone, or a wireless microphone (Urbanski Col. 3 line 65 – Col. 4 line 4).

Further, Urbanski teaches acoustic noise suppression in a speech communication system generally serves the purpose of improving the overall quality of the desired audio signal by filtering environmental background noise from the desired speech signal. This speech enhancement process is particularly necessary in environments having abnormally high levels of ambient background noise, such as an aircraft, a moving vehicle, or a noisy factory. One noise suppression technique is a spectral subtraction--or a spectral gain modification--technique. Using this approach, the audio input signal is divided into individual spectral bands by a bank of bandpass filters, and particular spectral bands are attenuated according to their noise energy

content. A spectral subtraction noise suppression prefilter utilizes an estimate of the background noise power spectral density to generate a signal-to-noise ratio (SNR) of the speech in each channel, which, in turn, is used to compute a gain factor for each individual channel. The gain factor is used as the attenuation for that particular spectral band. The channels are then attenuated and recombined to produce the noise-suppressed output waveform. In specialized applications involving relatively high background noise environments, most noise suppression techniques exhibit significant performance limitations. One example of such an application is the vehicle speakerphone option to a cellular mobile radio telephone system, which provides hands-free operation for the automobile driver. The mobile hands-free microphone is typically located at a greater distance from the user, such as being mounted overhead on the visor. The more distant microphone delivers a much poorer signal-to-noise ratio to the land-end party due to road and wind noise conditions. Although the received speech at the land-end is usually intelligible, continuous exposure to such background noise levels often increases listener fatigue. In rapidly-changing high noise environments, a severe low frequency noise flutter develops in the output speech signal which resembles a distant "jet engine roar" sound. This noise flutter is inherent in a spectral subtraction noise suppression system, since the individual channel gain parameters are continuously being updated in response to the changing background noise environment (Urbanski Col. 1 lines 20-63 & Fig. 3 bands).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Andrea in view of Soli and Suzuki to

incorporate received-speech-level measurement means for measuring a level of the received-speech signal at each predetermined frequency band, wherein the background sound level measurement means measures the level of the background sound in each predetermined frequency band and the received-speech clarifying means performs loudness compensation in which the gain for the received-speech signal is adjusted in each predetermined frequency band such that received speech output by the speaker is heard at almost the same intensity in the human auditory sense irrespective of the level of the background sound, and the resultant signal is output to the speaker as the received speech as taught by Urbanski to allow for gain parameters that are continuously being updated in response to the changing background noise environment in various frequency bands, wherein a user can operate said communication device in a noisy environment such as an automobile and still hear the speaker (Urbanski Col. 1 lines 20-63 & Fig. 3).

## *Conclusion*

8. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not

mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Michael C Colucci/
Examiner, Art Unit 2626
Patent Examiner
AU 2626
(571)-270-1847
Examiner FAX: (571)-270-2847
Michael.Colucci@uspto.gov

/Richemond Dorvil/
Supervisory Patent Examiner, Art Unit 2626